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2100 Pennsylvania Avenue, N. W.  
Washington, DC 20037-3202

EXAMINER

WOZNIAK, JAMES S

ART UNIT	PAPER NUMBER
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2655

DATE MAILED: 02/17/2004

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Please find below and/or attached an Office communication concerning this application or proceeding.

8

# Office Action Summary

Application No.

09/699,435

Applicant(s)

MURASHIMA, ATSUSHI

Examiner

James S. Wozniak

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-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

## Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133).
- Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

## Status

- 1) ☒ Responsive to communication(s) filed on 10/31/2000.
- 2a) ☐ This action is **FINAL**. 2b) ☒ This action is non-final.
- 3) ☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

## Disposition of Claims

- 4) ☒ Claim(s) 1-46 is/are pending in the application.
- 4a) Of the above claim(s) \_\_\_\_\_ is/are withdrawn from consideration.
- 5) ☒ Claim(s) 41, 42, 45, 46 is/are allowed.
- 6) ☒ Claim(s) 1-40 is/are rejected.
- 7) ☒ Claim(s) 43 and 44 is/are objected to.
- 8) ☐ Claim(s) \_\_\_\_\_ are subject to restriction and/or election requirement.

## Application Papers

- 9) ☒ The specification is objected to by the Examiner.
- 10) ☒ The drawing(s) filed on 31 October 2000 is/are: a) ☒ accepted or b) ☐ objected to by the Examiner.
- Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).
- Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).
- 11) ☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

## Priority under 35 U.S.C. §§ 119 and 120

- 12) ☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).
- a) ☒ All b) ☐ Some \* c) ☐ None of:
1. ☒ Certified copies of the priority documents have been received.
2. ☐ Certified copies of the priority documents have been received in Application No. \_\_\_\_\_.
3. ☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).
- \* See the attached detailed Office action for a list of the certified copies not received.
- 13) ☐ Acknowledgment is made of a claim for domestic priority under 35 U.S.C. § 119(e) (to a provisional application) since a specific reference was included in the first sentence of the specification or in an Application Data Sheet. 37 CFR 1.78.
- a) ☐ The translation of the foreign language provisional application has been received.
- 14) ☐ Acknowledgment is made of a claim for domestic priority under 35 U.S.C. §§ 120 and/or 121 since a specific reference was included in the first sentence of the specification or in an Application Data Sheet. 37 CFR 1.78.

## Attachment(s)

- 1) ☒ Notice of References Cited (PTO-892)
- 2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
- 3) ☒ Information Disclosure Statement(s) (PTO-1449) Paper No(s) 2, 4.
- 4) ☐ Interview Summary (PTO-413) Paper No(s). \_\_\_\_\_.
- 5) ☐ Notice of Informal Patent Application (PTO-152)
- 6) ☐ Other: \_\_\_\_\_.

**Detailed Action**

***Specification***

1. The disclosure is objected to because of the following terminological informalities: p.35 identifies the “unvoiced” in “voiced/unvoiced identification” with “noise,” while “speech mode decision” on p. 36 seems to be the voiced/unvoiced/noise discrimination of p. 46. In the speech art, “noise” is non-speech information, while “unvoiced” is speech data without a pitch structure and “voiced” is speech data containing a pitch structure. So, the examiner has interpreted said “unvoiced” to mean non-speech or noise and “voiced” to mean speech data, while the detection of “voiced” versus “unvoiced” speech data has been interpreted as occurring in the “speech mode” determination. The specification should be corrected to not use “unvoiced” and “voiced” as pertaining to voice activity detection (rather than different types of speech). Similar terminological confusion is present in Claims 3, 17, 28, 43, and 44.

Appropriate terminological correction is required.

2. The lengthy specification has not been checked to the extent necessary to determine the presence of all possible minor errors. Applicant's cooperation is requested in correcting any errors of which applicant may become aware in the specification.

***Claim Objections***

3. Claims 3 and 28 are objected to because of the following terminological informalities:

“voiced” information should be more clearly listed as --speech—information, as in the specification objection above. The examiner has interpreted “voiced” to mean speech for the application of prior art.

4. Claims 17, 43, and 44 are objected to because of the following informalities: “voiced” information should be more clearly listed as --speech-- information, while “unvoiced” information should be more clearly denoted as --non-speech-- or --noise—information, as in the specification objection above. The examiner has interpreted “voiced” to mean speech and “unvoiced” to mean noise or non-speech for the application of prior art.

5. Claims 27-40 are objected to because of the following informalities: “processing of performing, limiting etc” should be corrected to read --process for performing, limiting, etc” in Claims 27-29 and 38-40 and “program for processing of representing, dividing, switching, etc” should be corrected to read --containing processing for representing, dividing, switching, etc-- in Claims 30-37.

Appropriate correction is required.

***Claim Rejections - 35 USC § 103***

6. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

(a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person

having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negated by the manner in which the invention was made.

7. **Claims 1-40** are rejected under 35 U.S.C. 103(a) as being unpatentable over Gao (*U.S. Patent: 6,507,814*) in view of Jarvinen et al (*U.S. Patent: 5,960,389*).

With respect to **Claims 1, 15, and 27**, Gao discloses:

A speech signal decoding method, apparatus, and computer program (*decoder featuring program and data ROM, Col. 40, Lines 1-6*) for decoding information concerning at least a sound source signal, gain and linear prediction coefficients from a received signal, generating an excitation signal and linear prediction coefficients from decoded information, and driving a filter, which is constituted by the linear prediction coefficients, by the excitation signal to thereby decode a speech signal, comprising:

A first step of smoothing the gain using a past value of the gain (*Fig. 4, Element 403*);

A third step of decoding the speech signal using the smoothed gain (*reproduced speech signal, Fig. 5, Element 539*).

Gao does not specifically teach the limiting of a smoothed gain based on a fluctuation amount from the gain and smoothed gain, however Jarvinen discloses:

A second step of limiting the value of the smoothed gain based upon an amount of fluctuation calculated from the gain and the smoothed gain (*comparing the difference between an excitation gain and an excitation gain median to a threshold and replacing a gain value exceeding the threshold, Col. 10, Lines 21-26*).

Gao and Jarvinen are analogous art because they are from a similar field of endeavor in speech analysis through synthesis. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to combine the method and device for limiting excitation

gain in relation to exceeding a threshold as taught by Jarvinen with the speech decoder featuring gain smoothing to decode a speech signal as taught by Gao to prevent amplification of a random noise spike within a speech signal that would prove undesirable to the listener upon synthesis. It also would have been obvious to one of ordinary skill in the art, at the time of invention, to replace the median value used in threshold comparison as taught by Jarvinen with the average or smoothed value as taught by Gao, since both values with fall would fall within a middle range of a set of gain values and thus would be appropriate as gain replacement values so as not to exceed a threshold. Therefore, it would have been obvious to combine Jarvinen and Gao for the benefit of obtaining a speech decoder capable of limiting gain so as to prevent amplification of a random noise impulse, to obtain the invention as specified in Claims 1 and 15.

With respect to **Claims 2 and 28**, Gao discloses:

A speech signal decoding method, apparatus, and computer program (*decoder featuring program and data ROM, Col. 40, Lines 1-6*) for decoding information concerning at least a sound source signal, gain and linear prediction coefficients from a received signal, generating an excitation signal and linear prediction coefficients from decoded information, and driving a filter, which is constituted by the linear prediction coefficients, by the excitation signal to thereby decode a speech signal, comprising:

Deriving a norm of the excitation signal at regular intervals (*selection of an excitation signal from a codebook and an associated gain which normalizes the excitation signal, Col. 5, Lines 57-60*);

Smoothing the norm using a past value of the norm (*Fig. 4, Element 403. Also, it would have been obvious to one of ordinary skill in the art, at the time of the invention, that an*

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*averaging (smoothing) operation would include past values in order to smooth the gain over time to prevent excessive gain that would prove undesirable to the listener upon speech synthesis.);*

Changing the amplitude of the excitation signal in said intervals using said norm and the norm that has been smoothed (*smoothed gain applied to an excitation signal, Col. 6, Lines 45-55*); and

Driving the filter with the excitation signal, the amplitude of which has been changed (*excitation signal driving a synthesis filter, Col. 7, Lines 26-28, and Fig. 5, Element 531*).

Gao does not teach limiting the value of the smoothed norm based upon an amount of fluctuation calculated from the norm and the smoothed norm, however Jarvinen recites:

Limiting the value of the smoothed norm based upon an amount of fluctuation calculated from the norm and the smoothed norm (*comparing the difference between an excitation gain and an excitation gain median to a threshold and replacing a gain value exceeding the threshold, Col. 10, Lines 21-26*);

Gao and Jarvinen are analogous art because they are from a similar field of endeavor in speech analysis through synthesis. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to combine the method and device for limiting excitation gain (norm) in relation to exceeding a threshold as taught by Jarvinen with the speech decoder featuring gain smoothing to decode a speech signal as taught by Gao to prevent amplification of a random noise spike within a speech signal that would prove undesirable to the listener upon synthesis. It also would have been obvious to one of ordinary skill in the art, at the time of invention, to replace the median value used in threshold comparison as taught by Jarvinen with

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the average or smoothed value as taught by Gao, since both values would fall within a middle range of a set of gain values and thus would be appropriate as gain replacement values so as not to exceed a threshold. Therefore, it would have been obvious to combine Jarvinen and Gao for the benefit of obtaining a speech decoder capable of limiting gain (norm) so as to prevent amplification of a random noise impulse, to obtain the invention as specified in Claims 2 and 28.

With respect to **Claims 3 and 29**, Gao teaches the decoding method, apparatus and computer program involving excitation signal smoothing and limiting as applied to Claims 2 and 28. Also, Gao further discloses:

A first step of identifying a “voiced” (speech) segment and a noise segment with regard to the received signal using the decoded information (*classifying noise, voiced, and unvoiced speech for appropriate modeling of input speech information, Col. 4, Lines 23-30, for use in the speech decoding process as applied to Claim 2*).

Gao does not teach limiting the value of the smoothed norm based upon an amount of fluctuation calculated from the norm and the smoothed norm, however Jarvinen recites:

Limiting the value of the smoothed norm based upon an amount of fluctuation calculated from the norm and the smoothed norm (*comparing the difference between an excitation gain and an excitation gain median to a threshold and replacing a gain value exceeding the threshold, Col. 10, Lines 21-26*);

Gao and Jarvinen are analogous art because they are from a similar field of endeavor in speech analysis through synthesis. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to combine the method and device for limiting excitation gain (norm) in relation to exceeding a threshold as taught by Jarvinen with the speech decoder

featuring gain smoothing to decode a speech or noise signal as taught by Gao to prevent amplification of a random noise spike that would prove undesirable to the listener upon synthesis. It also would have been obvious to one of ordinary skill in the art, at the time of invention, to replace the median value used in threshold comparison as taught by Jarvinen with the average or smoothed value as taught by Gao, since both values would fall within a middle range of a set of gain values and thus would be appropriate as gain replacement values so as not to exceed a threshold. Therefore, it would have been obvious to combine Jarvinen and Gao for the benefit of obtaining a speech decoder capable of limiting gain (norm) so as to prevent amplification of a random noise impulse, to obtain the invention as specified in Claims 3 and 29.

With respect to **Claims 4, 18, and 30**, Jarvinen further discloses the method and apparatus for gain limitation as applied to Claims 1 and 15, in which the gain cannot exceed a predetermined threshold due to its replacement with an averaged excitation gain (*comparing the difference between an excitation gain and an averaged excitation gain to a threshold and replacing a gain value exceeding the threshold, Col. 10, Lines 21-26*). It also would have been obvious to one of ordinary skill in the art, at the time of invention, to utilize the gain-normalized absolute value of the difference to determine the relative “percent” by which a threshold is exceeded in regards to a lower limit and upper limit, because exceeding a lower threshold requires going below a threshold, yielding in a negative result, while the opposite is true for an upper threshold and a percentage threshold tracks the logarithmic properties of the loudness as heard by the human ear. Thus, in order to determine the absolute amount by which a threshold has been exceeded, it would have been obvious to one of ordinary skill in the art, at the time of invention, to implement an absolute value of the percent difference in the gain limiting process.

With respect to **Claims 5 and 31**, Gao does not teach a fluctuation amount calculation for fluctuation limitation represented by dividing an absolute value of a difference between the norm and the smoothed norm by the norm, but Jarvinen further discloses:

The amount of fluctuation is represented by dividing an absolute value of a difference between the norm and the smoothed norm by the norm, and the value of the smoothed norm is limited in such a manner that the amount of fluctuation will not exceed a predetermined threshold value (*comparing the difference between an excitation gain (norm) and an excitation gain (norm) median to a threshold and replacing a gain value exceeding the threshold, Col. 10, Lines 21-26*);

Also, it would have been obvious to one of ordinary skill in the art, at the time of invention, to utilize the normalized absolute value of the difference to determine the amount by which a percentage threshold is exceeded in regards to a lower limit and upper limit, because exceeding a lower threshold requires going below a threshold, yielding in a negative result, while the opposite is true for an upper threshold and a percentage threshold tracks the logarithmic properties of the loudness as heard by the human ear. Thus, in order to determine the absolute amount by which a threshold has been exceeded, it would have been obvious to one of ordinary skill in the art, at the time of invention, to implement such a normalized absolute value of the difference between an original and smoothed norm in the gain limiting process.

With respect to **Claims 6 and 32**, Gao does not teach a fluctuation amount calculation for fluctuation limitation represented by dividing an absolute value of a difference between the norm and the smoothed norm by the norm, but Jarvinen further discloses the decoder featuring the gain fluctuation limitation method as applied to Claims 5 and 31, above.

With respect to **Claims 7 and 33**, Gao further discloses:

The excitation signal in said intervals is divided by the norm in said intervals and the quotient is multiplied by the smoothed norm in said intervals to thereby change the amplitude of the excitation signal (*selection of an excitation signal from a codebook and an associated gain which normalizes the excitation signal, Col. 5, Lines 57-60, and smoothed gain applied to an excitation signal, Col. 6, Lines 45-55*).

With respect to **Claim 8 and 34**, Gao further discloses the excitation signal scaling method as discussed for Claims 7 and 33, above.

With respect to **Claims 9, 21, and 35**, Gao discloses the decoding method and apparatus utilizing gain smoothing and scaling as applied to Claims 1, 15, and 27. Gao does not teach switching between a scaled and original gain signal by utilizing a control signal, however Jarvinen discloses:

Switching between use of the gain and use of the smoothed gain is performed in accordance with an entered switching control signal when the speech signal is decoded (*selector, Fig. 4, Element 307, for switching between a gain and replacement gain values to implement the gain replacement method as applied to Claim 1 Col. 10, Lines 37-47*).

Gao and Jarvinen are analogous art because they are from a similar field of endeavor in speech analysis through synthesis. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to combine the ability to switch between an original and scaled excitation signal through the use of a control signal as taught by Jarvinen with the decoding method and apparatus utilizing gain smoothing and scaling as taught by Gao in order to provide appropriate system response of an original norm in the event of a noise spike and a

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scaled norm in response to a valid speech signal. Therefore, it would have been obvious to combine Jarvinen and Gao for the benefit of obtaining a speech decoder capable of selecting an appropriate excitation-scaling configuration through switching means to prevent undesirable auditory outputs, to obtain the invention as specified in Claims 9, 21, and 35.

With respect to **Claims 10 and 36**, Gao discloses the decoding method and apparatus utilizing norm smoothing and scaling as applied to Claim 2. Gao does not teach switching between a scaled and original excitation signal by utilizing a control signal, however Jarvinen discloses:

Switching between use of the excitation signal and use of the excitation signal the amplitude of which has been changed is performed in accordance with an entered switching control signal when the speech signal is decoded (*selecting between excitation vectors controlled by a threshold block, Col. 11, Lines 44-54*).

Gao and Jarvinen are analogous art because they are from a similar field of endeavor in speech analysis through synthesis. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to combine the ability to switch between an original and scaled excitation signal through the use of a control signal as taught by Jarvinen with the decoding method and apparatus utilizing excitation signal smoothing and scaling as taught by Gao in order to provide appropriate system response of an original norm in the event of a noise spike and a scaled norm in response to a valid speech signal. Therefore, it would have been obvious to combine Jarvinen and Gao for the benefit of obtaining a speech decoder capable of selecting an appropriate excitation-scaling configuration through switching means to prevent undesirable auditory outputs, to obtain the invention as specified in Claims 10 and 36.

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With respect to **Claims 11 and 37**, Gao in view of Jarvinen disclose the decoder featuring norm smoothing and limiting and a switching means to select an original or scaled excitation as applied to Claims 10 and 36.

With respect to **Claims 12, 24, and 38**, Gao adds:

Encoding an input speech signal by expressing the input speech signal by an excitation signal and linear prediction coefficients (*encoder using excitation vectors and LPCs, Col. 5, Line 64- Col. 6, Line 3*).

With respect to **Claims 13 and 39**, Gao further discloses:

Encoding an input speech signal by expressing the input speech signal by an excitation signal and linear prediction coefficients (*Col. 5, Lines 64- Col. 6, Line 4*).

With respect to **Claims 14 and 40**, Gao further discloses encoding method as applied to Claims 13 and 39.

**Claims 16 and 17** recite similar subject matter to Claims 2 and 3, respectively and thus are rejected for the same reasons.

**Claims 19 and 22** recite similar subject matter to Claims 5 and 10, respectively and thus are rejected for the same reasons.

**Claims 20 and 23** recite similar subject matter to Claims 6 and 11, respectively and thus are rejected for the same reasons.

**Claim 25** recites similar subject matter to Claim 13, and thus is rejected for the same reasons.

**Claim 26** recites similar subject matter to Claim 14, and thus is rejected for the same reasons.

*Allowable Subject Matter*

8. **Claims 41-46** are allowed.

9. The following is an examiner's statement of reasons for allowance. Prior art does not teach, nor fairly suggest:

- A smoothing coefficient calculation circuit that calculates an LSP average in combination with a decoder apparatus containing an LSP conversion circuit, gain smoothing circuitry which utilizes the LSP average, and a smoothing quantity limiter, with respect to **Claim 41**.

**Claims 42-46** are allowable as they further limit their parent claims.

Any comments considered necessary by applicant must be submitted no later than the payment of the issue fee and, to avoid processing delays, should preferably accompany the issue fee. Such submissions should be clearly labeled "Comments on Statement of Reasons for Allowance."

*Conclusion*

10. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure:

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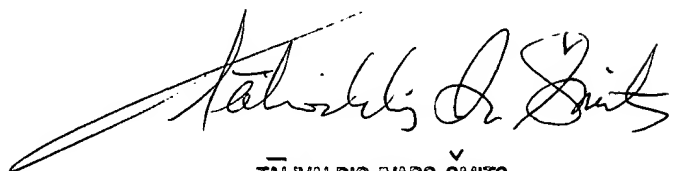
- Oshikiri et al (*U.S. Patent: 6,202, 046*)- teaches a method of classifying background noise and speech featuring gain smoothing and speech windowing using a predetermined frame length.
- Ertem et al (*U.S. Patent: 6,453,289*)- discloses a speech coder for noise reduction containing a VAD and adaptive gain limits.
- Copperi (*"Efficient Excitation Modeling in a Low Bit-Rate CELP Coder"*)- teaches a coder that utilizes excitation smoothing and windowing and an adaptive gain based on past gain values in decoding a speech signal.
- Kroon et al (*U.S. Patent: 5,664,055*)- teaches a CELP coder utilizing adaptive gain and gain limiting.

11. Any inquiry concerning this communication or earlier communications from the examiner should be directed to James S. Wozniak whose telephone number is (703) 305-8669 and email is James.Wozniak@uspto.gov. The examiner can normally be reached on Mondays-Fridays, 8:30-4:30.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Talivaldis Ivars Smits can be reached at (703) 306-3011. The fax/phone number for the Technology Center 2600 where this application is assigned is (703) 872-9306.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the technology center receptionist whose telephone number is (703) 306-0377.

James S. Wozniak  
2/11/2004



TĀLIVALDIS IVARS SMITS  
PRIMARY EXAMINER